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ON THE DESIGN OF LOUDSPEAKERS FOR BROADCAST MONITORING

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Research Department, Engineering Division
THE BRITISH BROADCASTING CORPORATION

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Summary

By designing its own monitoring loudspeakers for about the past forty years, the BBC has achieved a degree of continuity and consensus in a subject where there are almost as many opinions as there are designers. This Report describes the approach that has evolved within the Corporation towards design and assessment, and indicates how calculation and objective measurement are supplemented by experience and subjective judgement to arrive at a design that is acceptable to users in the broadcasting service. Each component of the loudspeaker is considered, and the problems of achieving consistency and reliability are addressed.

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LIST OF SYMBOLS

Symbol	Description	MKS Units
a	Effective radius of diaphragm	m
B	Magnet flux density	T
c	Speed of sound	345 ms^{-1}
C_{ab}	Acoustic compliance of box	N^{-1} m^5
C_{ms}	Mechanical compliance of drive unit	m N^{-1}
E_g	Input voltage to voice coil	V
f	Frequency	Hz
k	Wave number $2\pi f/c$	
l	Length of wire in coil	m
L_r	Drive unit/vent mutual radiation impedance	kg
M_{av}	Acoustic mass of air in vent	kg m^{-4}
M_{ms}	Effective moving mass of drive unit	kg
R_1	Acoustic resistance, box leakage	N m^{-5} s
R_2	Acoustic resistance, air damping material	N m^{-5} s
R_3	Acoustic resistance, vent viscous loss	N m^{-5} s
R_a	Acoustic radiation resistance	N m^{-5} s
R_{as}	Acoustic resistance of drive unit (due to mechanical losses)	N m^{-5} s
R_e	Electrical resistance of coil	Ω
s_d	Effective area of diaphragm	m^2
U_b	Total box volume velocity	m^3 s^{-1}
U_s	Volume velocity of diaphragm	m^3 s^{-1}
W_{ao}	Radiated power output	W
z_{ls}	Electrical impedance of drive unit in box	Ω
z_d	Electrical output impedance of filter network	Ω
Δ	Relative damping factor (see Section 8.3)	
η	Electroacoustic efficiency	
ρ	Density of air	1.18 kg m^{-3}

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1. INTRODUCTION

During its history, the BBC has found it beneficial to develop and sometimes manufacture a wide range of technical equipment. Over the years, as the range and suitability of commercially available equipment has increased, design and manufacture within the BBC has correspondingly decreased. As an example, BBC work on microphone design ceased about thirty years ago (although two microphones of BBC design are still commercially available).

The fact that loudspeakers are still the subject both of practical development and of fundamental research in this Department is a clear indication that commercial loudspeaker designs do not as yet meet the Corporation's requirements for broadcast monitoring. Such a view requires justification, and the reasons for it are outlined in Section 3, which also sets down the objectives and criteria underlying the design of a studio monitor. Section 2 is by way of a digression on the nature and definition of sound quality.

The remainder of this Report describes in some detail the design of the essential component parts, and their combination to form an acceptable loudspeaker assembly. To this day, the performance parameters that would ensure the acceptability of a loudspeaker cannot be completely specified, and are therefore also only partially within the realms of measurement. Given this primitive state of knowledge, where the final assessment can be made only by listening, the term 'design' as used in this Report must be understood to apply rather loosely in some contexts.

Finally, some methods of measuring loudspeaker behaviour and performance are reviewed. These represent landmarks and frontiers in the long-standing attempt to quantify subjective audio quality — an attempt which spans the lifetime of this author as well as that of loudspeaker development within the BBC.

2. ON THE QUALITY OF REPRODUCED SOUND

The importance of reproduced sound quality has been recognised for a long time; probably since Edison's invention of the phonograph. Certainly in their classic paper of 1925[1], Rice and Kellogg regarded the minimisation of non-linear distortion as a key factor in their contribution to amplifier and loudspeaker design.

Since then, the exact nature of some deficiencies in subjective quality has been determined, while the exact nature of others is as much of a mystery as ever. As a result, much has been written and believed which is not altogether based on repeatable observation or attributable to rigorously derived theoretical explanation. It is not the author's intention to enter this arena, but rather to try to describe the aspects of subjective tonal quality that have emerged as significant in balancing sound programmes during the BBC's accumulated experience of sixty years. It must however be said that during this time, no correlation has as yet been discovered between perceived sound quality and paranormal phenomena, cosmic radiation, or the oxygen content of any part of the considerable length of copper wire present in a typical broadcast chain.

Much of the music and speech broadcast by the BBC is intended to reach the listener sounding as nearly as possible as it did in the studio or auditorium of its origination. (Exceptions include sound effects, and some pop and light orchestral music which is partly created at the mixing desk.) There is therefore ample opportunity both for those who design and those who use monitoring loudspeakers to compare reproduced with original sound. Apart from the obvious advantage of this to the designer, who benefits both from his own listening and from the guidance of colleagues who are specialist listeners, this facility of comparison permits the users to assess and therefore at least partially allow for any characteristic defects of the loudspeakers.

Audible impairments can originate anywhere in the chain from microphone to loudspeaker, and fall into two broad categories: those caused by transducers, and those caused by the rest of the chain. Compared with the impairment generated by even a good loudspeaker, that introduced by properly designed electronics is always negligible; analogue sound recorders are perceptibly imperfect, but for critical applications, digital recorders have been available in the BBC since 1974[2]. The remaining component, the microphone, is certainly capable of influencing perceived sound quality, and one type can readily be distinguished from another by those skilled in programme balancing. Such people, however, rarely use the word 'impairment' in describing the difference between microphones, and can distinguish without difficulty the peculiarities of a known microphone type from those of a known loudspeaker type.

The users of monitoring speakers, then, are

highly practised in the assessment of quality, and are often able to identify by ear with astonishing consistency different microphones, loudspeakers, auditoria, orchestras, conductors, and the handiwork of colleagues at the mixing desk. They form a valuable pool of expertise, and the loudspeaker designer would be foolish to ignore their criticisms of his work, however improbable these occasionally may seem. 'Improbability' in this connotation usually denotes conflict with the designer's preconceptions rather than with the laws of physics.

The requirement that a loudspeaker should sound exactly like the original programme source is obviously unrealistic. To re-create the original sound field at every instant of time in a room of different size and acoustic properties is clearly not possible using a finite number of sound sources, and will certainly not be achieved by two signal channels terminating in two boxes, whatever each box contains. Indeed, a recording of a person speaking in a free-field room, replayed via a single loudspeaker behind an acoustically transparent curtain in a listening room, compared with speech in the listening room from the person himself, standing close to the loudspeaker, rarely deceives the listener. This impossibility of recreating the original sound even for a single source means that every loudspeaker user must evolve for himself a limited subset of 'reality' by which he attempts to judge the absolute quality of a loudspeaker. Small wonder, surely, that there is so little agreement between objective measurement and subjective judgement if there are as many subjective judgements of a loudspeaker as there are listeners. In fact, those whose work involves listening tend to influence and learn from each other, so that, in the BBC for example, there is a very considerable degree of consensus among loudspeaker users as to what constitutes high-quality sound, and in what way the output of a particular loudspeaker deviates from this ideal. This makes the job of designing broadcast monitoring loudspeakers easier in some respects than that of designing for the world in general.

To the author, whose work involves many activities besides critical listening, it often seems, after a break such as an annual holiday, that the peak of reproduced realism (as achieved within the BBC, at any rate), is a mere travesty of the original sound. This is probably true; the synthesis of a sound stage from a pair of loudspeakers is a game whose rules must be learnt, and which can be temporarily forgotten. Furthermore, the rules are not universal, but differ by an unknown amount from one listener to another.

To summarise: this section is not intended to convey a doctrine of despair. Rather, it draws attention to the fact that while subjective judgements of loudspeakers can never be absolute, sufficient consensus exists among loudspeaker users in at least one broadcasting organisation to assure the designer of a highly coherent reaction to his efforts.

3. DESIGN OBJECTIVES AND CRITERIA

3.1 Reasons for in-house design

By numerical superiority, the domestic consumer represents by far the largest share of the loudspeaker market, in Britain and elsewhere; second is the recording industry, with broadcasters a rather poor third.

The three types of user also differ in their requirements. The domestic 'hi-fi' enthusiast is mostly in an extremely poor position to make critical comparisons under known conditions, and is probably guided mainly by the independent preferences of contributors to publications on the subject, taken in conjunction with current trends in manufacturers' claims.

The use of loudspeakers by recording studios is very similar to that by broadcasters, namely, to monitor, balance and assess programme output before it reaches the public. Requirements do however differ in several respects, at least as far as the BBC is concerned. The following are the most significant:

(1) The balance and recording of pop music forms a considerable part of most recording studios' output, but only a small part of the BBC's. Loudspeakers for pop monitoring are normally required to produce very high sound pressure levels so that material may be reproduced at some approach to its original level, especially for subsequent appraisal by its originators. To generate levels in excess of 120 dB at frequencies below 100 Hz requires very large cabinets and drive units, with powerful amplifiers, and would for most BBC applications be uneconomic in both cost and space.

(2) In total, the BBC uses some thousands of monitors, representing a large capital investment, and economics dictate that these must have a service life of ten to fifteen years. During this time it must be possible both to buy new speakers of a given type, and to service old ones; great value is therefore placed on the long-term availability and consistency of drive units.

(3) In a large broadcasting organisation, almost every member of staff who balances programmes has to work in a number of different

control areas; the greater the degree of subjective consistency between monitors, both of the same and of different types, the less difficult it is for a studio manager or sound supervisor (the titles used in Radio and Television respectively) to produce consistent sound balances. For this reason, there is a requirement not only for long-term consistency of a given loudspeaker type, but also for reasonable similarity between different types. This objective is difficult to achieve, and must not be allowed to prevent the replacement of obsolete types by perceptibly improved designs; in other words, some degree of engineering compromise is necessary.

Most present BBC requirements are met by three monitor types, of cabinet volumes 5.6, 28, and 109 litres (respectively about 0.2, 1, and 4 cubic feet). The smallest of these (LS3/5A) is of 'bookshelf' size[3]; the 28-litre monitor (LS5/9) is of about average domestic 'hi-fi' size[4], while the largest (LS5/8) is rather smaller than many commercial studio monitors[5].

3.2 Design criteria

A monitoring loudspeaker is based on design criteria which usually fall into two broad categories — *general*, which apply to all monitors; and *specific*, which distinguish one design from others. As applied within the BBC, these are as follows.

3.2.1 General criteria

(a) *Reliability*

In broadcasting, as indeed in most fields of application, the reliability of technical equipment is of paramount importance. As far as possible, equipment is designed to provide an adequate service life, and to withstand inadvertent misuse of any likely nature. Recent BBC monitor designs can withstand indefinite woofer overload, and short-term tweeter overload, provided that amplifiers of not more than the recommended power output are used. (The fact that *most* units are 'indestructible' does not of course preclude the failure of *some* units over a period of time in normal or abnormal use.) Economics necessitate a service life of about fifteen years, which means that (say) 75% of drive units should survive for this length of time, a failure rate of about 1.5% per annum. For a newly-developed unit, the probable lifetime is difficult to predict, but a good indication can be obtained from accelerated life tests, evaluated carefully in the light of past experience.

(b) *Consistency*

As far as the readily measurable parameters of a loudspeaker assembly are concerned, care in manufacture will usually ensure reasonable consistency, both in the short term and even over a period of years. High subjective quality, however, because it often cannot be related directly to materials or manufacturing methods, is not so easy to achieve even in the short term, and very great care is needed to maintain it over the product lifetime.

In a broadcasting organisation with a large number of studios, a high degree of subjective consistency between nominally identical loudspeakers is very desirable, because it allows a programme balancer to use any control room without having to acclimatise to the loudspeakers, for example by listening to a known recording. This requires long-term as well as short-term manufacturing consistency, so that new assemblies can be obtained and faulty ones refurbished. It is certainly impracticable to keep monitors together in matched stereo pairs, and in any case, in the event of a failure, immediate substitution of one monitor must be possible without impairment of the stereo performance. For commercial loudspeakers aimed at the domestic 'hi-fi' market, long-term consistency of this kind is not a requirement; indeed, to 'freeze' a design in that way would conflict with many manufacturers' policies of continual improvement.

In a range of studio monitors such as that designed by the BBC, another consideration is the degree of subjective consistency that can be achieved between the different types. In the development of a new monitor, similarity to existing units is borne in mind as a criterion, although this must not, of course, be allowed to prevent such improvements in quality as advances in technology may permit.

(c) *Sound quality*

The above discussion on consistency has been chiefly with respect to subjective sound quality. However, it is also important that quality be consistently high rather than consistently low. Some aspects of subjective quality have been touched upon in Section 2, and some attempts to measure it will be discussed in Section 9. Suffice it to reaffirm here that the requirement for reproduced quality as close as possible to the original sound is one of the main reasons for in-house design.

(d) *Interaction with surroundings*

The exact nature of a loudspeaker's interaction with a room is extremely complex, and no universally optimum properties have yet been convincingly postulated for the loudspeaker, the room, or their combined characteristics. From long experience, both in the BBC and elsewhere, some guiding principles have however emerged.

The most significant and obvious property of a loudspeaker that affects its behaviour in a room is its directionality; another is its sheer physical presence in the room, which evidently has more effect for large than for small cabinets. Experience has shown that a highly unidirectional unit can generate a well-defined stereo image only over a very small listening area, but that its tonal quality is reasonably independent of the nature of its immediate surroundings. Conversely, a unit that approximates to omnidirectionality has a wider listening window, but is tonally much more affected by the room acoustic. Yet different rules of thumb seem to govern a pressure-gradient loudspeaker, whose tonal quality appears noticeably affected by the acoustic properties of the surface behind it, presumably due to its bidirectionality.

Ideally, a control room or other listening area should have an excellent acoustic, special care being taken over any acoustic treatment placed in close proximity to the loudspeaker positions. In practice, a modern studio control room is required to accommodate a considerable quantity of equipment, including a large control desk, and to provide a good view of the studio through an observation window which may occupy most of one wall, with a reflection coefficient very close to unity. In a television sound control room, problems are made even worse by the provision of a bank of picture monitors.

In practice, the directionality of a normal direct-radiator loudspeaker in a cabinet is very difficult to control without seriously affecting tonal quality (e.g. by placing slots or lenses in front of drive units), except perhaps by choice of the number and size of drive units, and even this is often dictated by the required sound level and the permissible size of cabinet.

3.2.2 Specific criteria

(a) *Box size*

Because of the limitations on space in a control room, the overriding feature of any monitor is its size. The shape of the box must also be considered; in the author's experience, the influence of box shape on tonal quality is about an order of magnitude less than that of the drive units: in other words, changing the drive units (especially the low-frequency unit) has a far greater effect than changing the shape or size of the box. Users show a surprising degree of conservatism about box shape, and as this is uncritical except for the extreme box shapes used when wavelength-dependent effects are sought, recent BBC monitors (LS5/8 and LS5/9) have frontal dimensions approximating to the 'golden ratio' of 1.62:1, which seems to be preferred by almost all users. Cabinet design is considered in more detail in Section 6.

(b) *Power output and bandwidth*

The generation of high acoustic power at low frequencies requires the movement of large quantities of air, so that box size, power output, and low-frequency band limits are interdependent. Box size is such an overriding consideration that it has been treated separately, but for a given volume, some form of design compromise must always be reached between lower limiting frequency and maximum output level. Fortunately, small monitors are normally used in small control areas, where less acoustic power is needed to generate a given sound level, one of those rare applications in which physics favours the engineer.

Generally accepted broadcast bandwidths (−1 dB points) are 40 Hz to 15 kHz for FM radio, and 40 Hz to 14 kHz for television sound. Ideally, then, the output of all monitoring loudspeakers should extend downwards at least to 40 Hz; in fact, no BBC-designed monitor has ever maintained flatness to so low a frequency. The LS3/5A, LS5/9 and LS5/8 have half-power frequencies of 80, 56 and 50 Hz respectively, with respective mid-band SPL capabilities of 96, 108 and 114 dB; these figures represent a reasonable working compromise. If any future broadcasting standards require greater bandwidth, then of course this will have to be provided.

At the upper end of the spectrum, power output should be substantially flat to at least 15 kHz for radio monitoring; users in the television service in fact require to check programme output for absence of line whistle at 15.625 kHz.

3.3 General design approach

Within the BBC, good communication is maintained between designers and users of loudspeakers[6]; a new design emerges from either a new requirement, or an existing requirement that can no longer be economically fulfilled. Each projected design is considered from the viewpoint of the existing range: on rare occasions, a new loudspeaker type can replace more than one of those already in use.

Invariably, the first constraint on a new design is that of maximum box size; all other requirements have some degree of flexibility, however small. Given a box, the next decision is what to put inside it. Sometimes, at least one existing drive unit may be suitable; otherwise, two or more new drive units may be required. In practice, other work, including basic research, may well have pointed the way, and given rise to one or more prototype drive units good enough to form the basis of a new assembly.

For about the past fifteen years, it has been

possible to cover the range from 50 Hz to 15 kHz with only two drive units. Each additional unit involves such an increase in materials, weight, complexity, initial setting-up, and cost, that a new design incorporating more than two drive units would have to offer a very great advance indeed in subjective quality to be seriously considered.

The design begins, therefore, with a box, a great reluctance to put more than two drive units inside it, and perhaps at least one existing drive unit that may be suitable. The behaviour of low-frequency drive units is closely related to the properties of the box, and if a new unit has to be designed, the size of the box will greatly influence its design parameters.

The final major decision is whether to use a high-level (passive) or a low-level (usually active) filter arrangement for crossover and 'equalisation'. In this connotation, crossover denotes the band-splitting of power between drive units, and equalisation a (usually gentle) slope in electrical response-frequency characteristic designed to achieve approximate overall flatness of acoustic response within the design bandwidth. Careful listening tests on prototype LS5/9 monitors have indicated that there is very little subjective difference indeed between high-level and low-level filtering, so that considerations such as convenience and economy should chiefly determine the final choice.

4. LOUDSPEAKER DRIVE UNITS: GENERAL CONSIDERATIONS

4.1 The various motor mechanisms

The great majority of loudspeaker drive units now available are of the moving-coil electromagnetic type, including all of those that have ever been used by the BBC. Nevertheless, there are many possible kinds of motor mechanism, falling into two broad categories — electrostatic and electromagnetic; these are now briefly reviewed.

4.1.1 Ionic or 'corona wind' (Fig. 1)

This type of transducer acts on air molecules by ionising them, then exerting a force by applying an electric field. Of all loudspeaker types, this is the most direct-acting, having no moving parts other than the air itself. So far, it appears little more than a scientific curiosity, although it is believed that an assembly including an ionic tweeter is available in the USA. Proven practical advantages are not known to the author; reported disadvantages[7] are the necessity to provide very high polarising and drive voltages, and the generation of ozone (which is poisonous) and of noise by the ionising discharge.

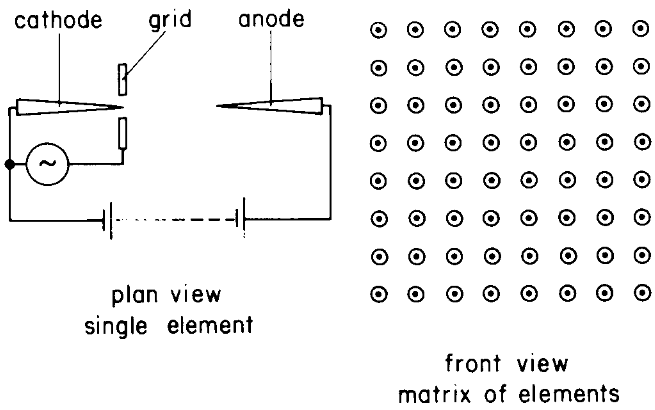

Fig. 1 - Ionic or 'corona wind' transducer, showing details of single element, and also matrix of elements all connected in parallel.

4.1.2 Electrostatic (Fig. 2)

Although this term, strictly speaking, includes ionic and piezoelectric devices, it is usually applied to transducers incorporating a very thin charged membrane of (slightly) conducting material suspended in an electrostatic field. Such devices, unlike the ionic type, cannot be freed from the properties and behaviour of available materials, but are almost as direct-acting in the sense that pressure is applied over an area of material which effectively weighs less than the air to which the pressure is directly transferred. In a well-known proprietary form[8,9], the electrostatic loudspeaker is capable of reproducing the full audio range at high quality; the diaphragm is exposed to the air on both sides, and therefore behaves as a doublet or pressure-gradient source with a figure-of-eight directional pattern[10]. The membrane is sandwiched between two

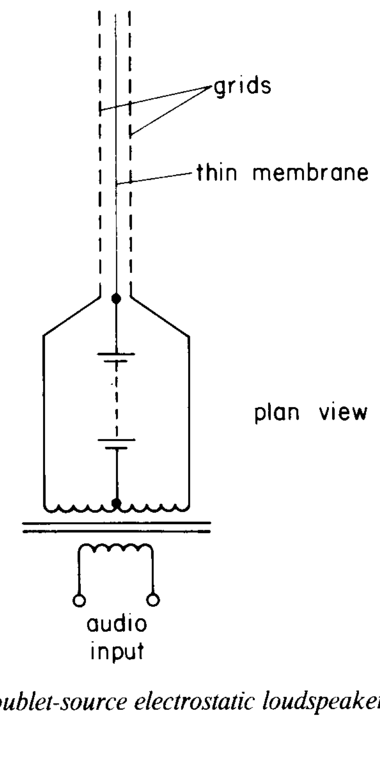

Fig. 2 - Doublet-source electrostatic loudspeaker (plan view).

grids which form the fixed electrodes, and is divided into separately driven areas for control of its radiation pattern at mid and high frequencies.

The disadvantages of bidirectionality in a typical control room have been touched upon in the previous section; a further disadvantage of the doublet source is its poor radiation efficiency, and consequent inability to generate sound pressure levels greatly exceeding 100 dB at low frequencies.

The possibility of building a wideband electrostatic driver into a cabinet has been considered. This might appear to solve the problems both of bidirectionality and of low-frequency inefficiency — it certainly does so for an electromagnetic bass unit (whose inability to cover the audio spectrum requires the addition of a tweeter). Such a hypothetical experiment will illustrate very well the primary difference in behaviour between an electromagnetic and an electrostatic driver. A typical electromagnetic unit generates large forces to move large masses, and at most frequencies is largely 'unaware' of air loading: in other words, the volume velocity it generates is almost independent of the opposing air pressure. In contrast, the diaphragm of an electrostatic unit has negligible mass at most frequencies, and its behaviour is governed almost entirely by the air pressure that opposes its movement. For a loudspeaker in a box to generate a given sound pressure level at a listener's ear requires an air pressure in the box which at low frequencies depends only on the volume of the box, the distance of the listener, and the room size and reverberation time. For example, the box pressure of an LS5/8 at 50 Hz is about 31 dB higher than that experienced by a listener 1.5 m distant in a typical listening room. The peak diaphragm pressure that can be generated by an electrostatic unit depends only on the dielectric strength of air; assuming this to be 3 MVm^{-1} (an ideal figure that could not be closely approached in practice), the peak output level (at the listener's position) would be 93 dB SPL, whereas an unboxed unit (doublet source) in similar conditions could generate about 110 dB SPL. Thus the use of a box, while perhaps solving problems of directionality, could result in a loss in output of about 17 dB at 50 Hz, this loss increasing with reduction in frequency at 12 dB per octave. In view of the fact that all of the levels quoted for an electrostatic driver are purely theoretical maxima, from which in practice it might be realistic to subtract about 10 dB, it is clear that an additional 17 dB loss could not be afforded, and that a boxed electrostatic driver of reasonable frontal area would at best be marginal in terms of low-frequency output power for studio monitoring.

The choice of drive motor must in practice always represent some compromise between theoretical excellence and practical applicability. The electrostatic drive unit has two disadvantages: the fundamental limitation on reproduced sound level, and the cost of providing a drive transformer and high polarising voltage.

4.1.3 Piezoelectric (Fig. 3)

Piezoelectricity has traditionally been associated with brittle substances such as Rochelle Salt and quartz, but during recent years there has been increasing interest in piezoelectric plastics[11], especially the most active of these, polyvinylidene fluoride (PVDF).

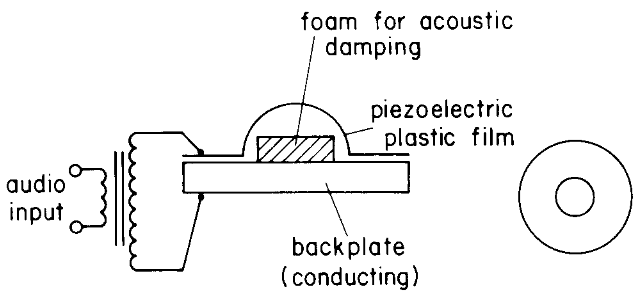

Fig. 3 - Piezoelectric dome tweeter.

As a motor mechanism for a loudspeaker driver, the piezoelectric effect is of all the least direct — it is the only arrangement that does not generate a force relative to a fixed reference (e.g. a magnet, frame, etc.). Rather, it causes a piece of material to alter its shape in the same way as by an applied force; indeed, by the application of a suitable force, its action may be cancelled. Thus by fixing one surface of the material and applying the appropriate electric field, another surface may be made to move by a distance depending on the piezoelectric coefficient and elastic modulus of the material, and the direction and magnitude of any opposing forces. To avoid electrical and mechanical damage to the material, strains must usually be considerably less than 1%. Large movements cannot therefore be obtained from very simple mechanical configurations, and for this reason the only successful loudspeaker applications have been in high-frequency drive units, where the required movements are small.

About six years ago, an investigation was undertaken at BBC Research Department into the feasibility of designing a usable piezoelectric tweeter[12]. Several interesting conclusions emerged, none of which suggested that such a unit could advantageously replace the moving-coil type currently in use.

First, the material (PVDF), if used in a simple configuration such as a dome, needs to be very thin (e.g. 100 microns) for reasonable unit sensitivity. It

must be vacuum-formed into a dome, then coated on both sides with a very thin film of a metal, such as aluminium, by vacuum evaporation; finally, polarisation (by which the piezoelectric properties are activated) is effected by the application of the maximum sustainable electric field between the surfaces for some time at about 100 °C. Even if strict laboratory cleanliness is observed during manufacture, the electrical integrity of such a thin sheet is difficult to preserve; any imperfections reduce the maximum applicable field.

Second, even if ideal performance could be realised, the device would be capable of lower acoustic output than a dynamic tweeter of comparable size. At low frequency, the input level is limited by the onset of dielectric breakdown, while at high frequency, dielectric loss, which appears to increase with temperature, can cause 'thermal runaway' and resultant dielectric breakdown, limiting even further the maximum safe input level. By using a much larger dome, adequate output power can be obtained, but at the expense of excessive directionality.

Third, PVDF is a stiff material with little internal damping, rather like the more familiar PVC (polyvinyl chloride) in its unplasticised form. A dome formed from a thin sheet exhibits a large number of resonance modes within the working frequency range of the unit. In the prototypes, many modes were both observable by optical interferometry, and audible in listening tests.

Fourth, the tweeter requires a matching transformer to operate with an ordinary power amplifier; this can be simple in design, but increases unit cost.

Since this work was undertaken, at least one piezoelectric plastic dome tweeter has been marketed. However, although the problems of manufacture have evidently been solved, the other disadvantages mentioned above appear to remain.

Finally, it should be mentioned that for some years, a different type of piezoelectric tweeter has been available. This uses a piezo crystal in the bending mode, acting as the motor to drive a diaphragm; some models are horn-loaded, presumably to maximise the acoustic output, and all are intended to be driven directly from a conventional power amplifier. These tweeters are cheap and extremely robust, but somewhat deficient in output below 5 kHz; none that the author has tested could be described as high quality.

It appears, therefore, that piezoelectric drivers have some way to go before they supplant electromagnetic units, even at the upper end of the audio spectrum. Considerable development, however, has taken place in the range and quality of commercially available piezoelectric tweeters, and this conclusion may have to be revised in the foreseeable future.

4.1.4 Dynamic (Fig. 4)

This term is used of both loudspeakers and microphones, and denotes a diaphragm driven by (or driving) a coil of wire in a magnetic field: hence the more accurate description of 'moving-coil'. The other application of electromagnetism, the ribbon or leaf, is considered separately in the next sub-section.

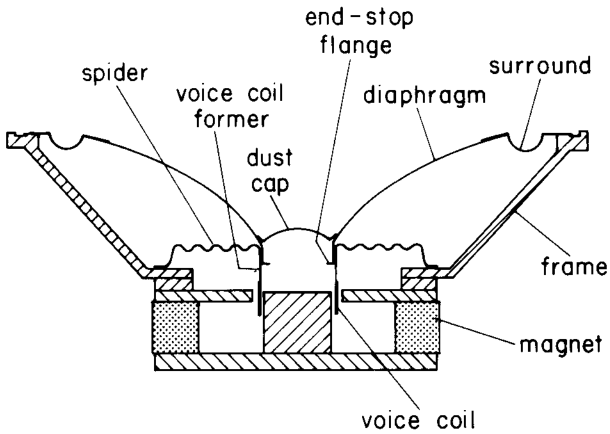

Fig. 4(a) - Moving-coil or 'dynamic' low frequency driver as used in the BBC monitors LS5/8 and LS5/9.

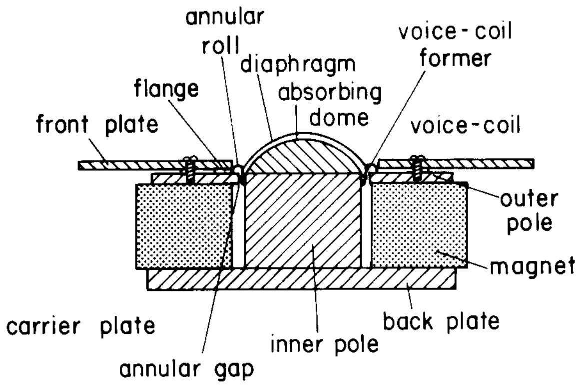

Fig. 4(b) - Moving-coil dome tweeter.

A current-carrying conductor in the field gap of a magnet experiences a force relative to the magnet; insofar as a force is generated relative to a fixed reference, the 'dynamic' and electrostatic motor mechanisms are very similar. The important practical difference is that while it is easy to set up an electric field occupying a large volume of space, a magnetic field can be generated only in the inevitably small volume bounded by the polepieces of a magnet assembly. To move the large volume of air required to generate high sound levels at low frequencies, the

moving-coil motor requires a diaphragm sufficiently rigid that it can be driven by a force distributed over only a small fraction of its area.

Dynamic drive units have two common forms of diaphragm, illustrated in Fig. 4 (a) and (b) respectively: a 'cone' (in practice more often a shallow horn shape), with its apex truncated and driven by a voice coil; and a dome, with its coil at the periphery. The detailed design of both types will be considered in later sections. The most common of all loudspeaker assemblies is probably a two-unit box containing a cone woofer and dome tweeter, with comparable coil diameters (the required power inputs being of the same order). All three current BBC monitors are of this type.

Since its introduction sixty years ago[1] the dynamic driver has dominated the loudspeaker market. Initially, it had few rivals; more recently, its continued market success has been due to its cheapness, robustness, and input impedance well suited to direct drive by cheaply realisable power amplifiers. In addition, many listeners (including the author) believe that at its best it offers sound quality not bettered by any other type of transducer. It is also capable of generating by far the highest sound levels, especially at low frequencies.

4.1.5 Ribbon and leaf types (Fig. 5)

A ribbon transducer, as its name implies, consists of a very thin conducting strip suspended between the polepieces of a magnet (see Fig. 5(a)). The whole radiating surface is in the magnetic field, which ideally would form a 'curtain' of flux parallel to the width of the ribbon and just thick enough to accommodate its maximum travel. Unlike the magnet of a dynamic driver, which has a narrow gap between polepieces of relatively large area, the magnet of a ribbon unit has a wide gap between polepieces of small area. Because of their mutual repulsion, the lines of flux crossing the gap do not of course form a curtain, but rather expand into a shape closer to cylindrical than planar. The ribbon only 'sees' a small part of the total flux, so that the magnetic circuit is largely ineffective in its intended function. To maximise magnetic efficiency, the ribbon, and therefore the polepiece gap, must be long and narrow.

It is thus not surprising that the ribbon transducer has been used chiefly as a microphone, which benefits from small size in all respects except noise performance. A powerful bass driver would require a magnet of quite unreasonable size and weight (about 0.5 tonne for the equivalent output power of an LS5/8 woofer); also, the stray magnetic field could present a variety of problems. However, ribbon tweeters are available if not common; they have three chief disadvantages. One is the length of the ribbon, which can lead to excessive directionality in one plane. The second is the effect of ribbon heating, which causes low conductivity in the centre, where the temperature is highest, resulting in maximum amplitude of movement at the edges, which reduces the power input at which ribbon rupture is likely to occur. The third is the very low electrical impedance of the ribbon, requiring the unit to be driven via a transformer.

The problem of non-uniform current distribution in a metal ribbon has led to the development of the 'leaf' transducer — again, like the ribbon unit, designed only as a tweeter[13]. In this the ribbon is replaced by a sheet of insulator carrying a conducting track arranged in a rectangular 'spiral' (see Fig. 5(b)). The two shorter sides of the rectangle are not subject to a magnetic field, but the longer sides, which of course carry current in opposite directions, are placed in opposing fields, so the 'leaf' approximates in behaviour to a ribbon, but without the possibility of temperature-dependent current distribution. A further advantage of the leaf is that it can be designed to

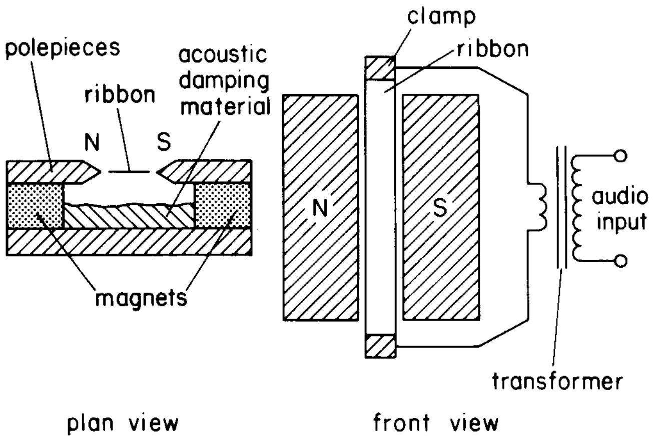

Fig. 5(a) - Ribbon tweeter.

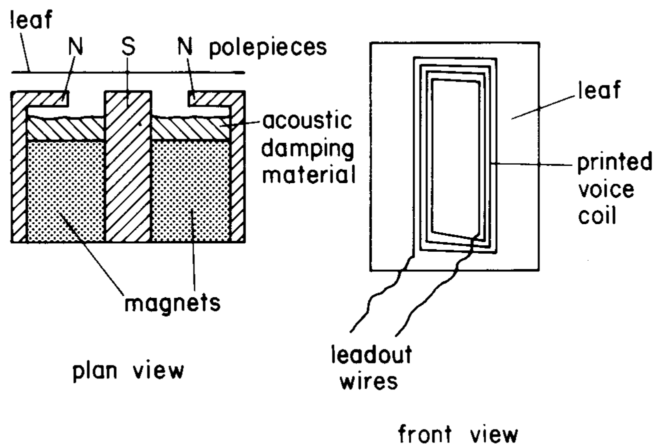

Fig. 5(b) - 'Leaf' tweeter.

present a convenient input impedance without the use of a transformer. Because there are two magnetic gaps with an intermediate polepiece, the leaf has to be placed, not in the gaps, but in front of the magnet, so that the magnetic circuit is even more inefficient than in a typical ribbon unit.

A feature of the leaf tweeter is its very low electrical inductance, which, combined with the direct way in which the leaf is driven (the coil being integral with the membrane), leads to high-frequency performance which can extend well beyond the audio band. For use in one-eighth scale modelling work, a loudspeaker designed by the author[14] incorporates a proprietary leaf tweeter providing output of excellent quality up to 100 kHz.

Both leaf and ribbon units are somewhat deficient in output power below about 5 kHz, which tends to preclude their use in two-unit loudspeaker designs.

One interesting variant on the leaf principle has recently been reported to the author. A membrane carrying a large array of conductors is placed close to one or between two corresponding arrays of magnets, and thus behaves as a doublet source, in the same way as the more familiar wide-range electrostatic driver. Whether, after development, such a transducer could rival the electrostatic type, is a matter open to speculation.

4.2 Horn loading

Although the benefit of a flared tube in increasing the loudness of a 'musical' wind instrument has been known since Biblical times, the earliest example of a horn-loaded loudspeaker is probably the phonograph, common during the latter half of the last century. The phonograph or gramophone illustrates very well the purpose of a horn as primarily employed then and since: namely, to obtain the maximum acoustic power output from a small mechanical or electrical power input by maximising radiation efficiency.

At the mention of horn loudspeakers, most people think of outdoor public-address systems. It is easy to forget that the design criteria for such transducers are *very different* to those for high-quality monitors, a fact which may account for the somewhat tarnished public image of the horn, and which may contribute to its neglect by most loudspeaker designers. Also, now that powerful amplifiers are cheaply available, high electroacoustic efficiency is no longer at a premium, although it could bring indirect advantages, e.g. in reliability, as a result of lower voice coil temperatures.

In fact, the horn and its many possible forms of flare have been the object of serious study since before the First World War[15,16], and several high-quality loudspeaker assemblies which include one or more horn drivers are currently available. It may be that the design of a horn unit involves more work than that of a direct radiator — as well as the horn itself, which can be of quite complex form, the driving transducer or 'compression driver' also requires care in design[17]. It may be, too, that such care would be well repaid, and that the horn deserves more general consideration in loudspeaker design.

5. THE DESIGN OF LOW-FREQUENCY DRIVE UNITS

5.1 Introduction

In this section, the design of low-frequency moving-coil or 'dynamic' drive units will be considered in some detail. As pointed out in Section 1, some aspects of loudspeaker design are calculable with reasonable accuracy, whereas others are essentially empirical. In fact, only at frequencies below about 400 Hz can the behaviour of a typical bass unit be predicted, in conjunction with the box containing it — which forms an essential part of the mechanoacoustic system.

The low-frequency physical behaviour of a bass unit in a box (whether vented or unvented) was possibly first analysed by Olson[18], but it was Thiele[19] who first pointed out that the complete system may to a good approximation be represented in the same way as an electrical low-pass filter. This realisation not only makes available a wealth of accumulated expertise in filter design, but also emphasises the interdependence of the drive unit and box to form a composite system. Thiele's model is readily manipulated by any microcomputer, and much may be learnt from it about the relationship between bass unit and cabinet before any plastic is moulded or wood sawn. In practice, the author has found that even with some simple refinements, the model does not exactly correspond to reality; but almost any comparison between acoustic theory and practice reveals similar discrepancies. In any case, the chief problems of the loudspeaker designer lie in the midrange rather than at the extremes of the audio-frequency band.

Examination of low-frequency systems shows that there is little to be gained by placing a small drive unit in a very large box, and even less by attempting to accommodate a large unit in a small box (practicable values of unit parameters such as moving mass and suspension compliance being assumed in both cases). Another useful conclusion is that the

fundamental mechanical resonance frequency of a bass unit should be below its intended lower band limit.

At mid frequencies, the efficiency of the unit is largely independent of the box, and is maximised by the use of a powerful magnet and a light diaphragm. The voice coil, assumed to be longer than the magnet gap (Fig. (4a)) should be designed to obtain the largest practicable mass of copper in the magnetic field, and the minimum amount outside it; in other words, the smaller the required peak-to-peak travel or 'throw' of the unit, the shorter will be the voice coil, and the higher the efficiency.

Towards the upper band limit of the unit (around 2 to 3 kHz for BBC designs) the efficiency is governed by both the mechanical behaviour of the moving parts (discussed later) and by the electrical inductance of the voice coil. This latter parameter is highly dependent on the amount of copper, and a short voice coil is therefore of especial benefit in extending upwards the useful bandwidth of a bass unit.

The relationship between bass unit and box is therefore important not only in achieving maximum low-frequency efficiency, but can also indirectly govern the performance at the upper band limit of the unit.

The other less readily quantifiable requirements of a low-frequency driver are its tonal quality (which usually dominates the tonal quality of the overall assembly) and its reliability. At the present state of knowledge, these must be approached from a chiefly empirical viewpoint.

5.2 Low-frequency behaviour: lumped-element model

It is not the purpose of this Report to reproduce the lucid and progressively more detailed bass-unit analyses of Novak[20], Thiele[19], Small[21] and others, but rather to provide an outline of the work, then point out some aspects of its application to practical loudspeaker design.

The model used in these analyses is of the 'acoustic impedance' type[10], in which pressure is represented by voltage, and volume velocity (cubic metres per second of air) by current.

A much simplified lumped-element acoustic representation of a bass driver in a vented box is shown in Fig. 6(a). The acoustic efficiency of a direct-radiator unit in a box is very low (about 2.5% at 300 Hz for the most efficient BBC monitor, the LS5/8), so that the radiation resistances of bass unit and vent are very small, and may be neglected in computing the mechanoacoustic behaviour of the system, which therefore lends itself to very simple circuit analysis. When the voice coil is driven by a power amplifier (a good approximation to a voltage source) the diaphragm generates a volume velocity given by

$$U_s = \frac{E_g Bl}{R_e s_d} \tag{5.1}$$

Because at low frequencies the distance between the bass unit and the vent is much less than one wavelength, the loudspeaker may be considered to

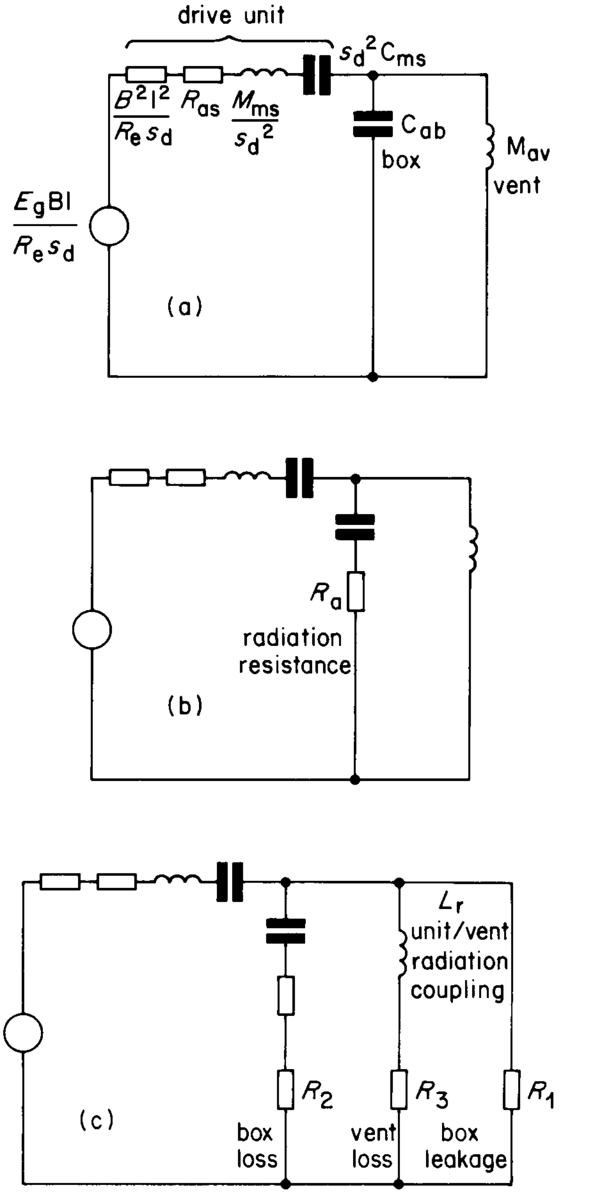

Fig. 6 - Lumped-element representations of bass unit in vented box, after Thiele.

(a) Simple model
(b) With radiation resistance added
(c) With box and vent losses added

'see' one common radiation resistance R_a driven by the total volume velocity flowing into or out of the box, as shown in Fig. 6(b). To a reasonable approximation[10], the overall radiation resistance at frequency f is given by

$$R_a = \frac{\pi \rho f^2}{c} \tag{5.2}$$

and the radiated power is

$$W_{ao} = U_b{}^2 R_a \tag{5.3}$$

It is then readily shown that the electroacoustic transfer function (ratio of fixed-distance free-field output sound pressure to input voltage) takes the form of a high-pass filter, fourth-order for a vented box and second-order for a closed box. The efficiency asymptotes with increasing frequency to

$$\eta = \frac{\rho B^2 l^2 s_d{}^2}{4\pi c R_e M_{ms}{}^2} \tag{5.4}$$

Thiele suggests the addition of further electrical filtering in the audio signal path so that the loudspeaker becomes part of a higher-order high-pass 'network', an approach which has since been developed and described in a number of papers. The author has some reservations about such extensions of the model, for two reasons. First, steep rates of bass roll-off tend to cause ringing at the cutoff frequency, which is detected and disliked by users, who refer to 'loose' or 'sloppy' bass. Second, the simple models of Fig. 6(a) and (b) do not, in general, accord very closely with measured performance. Their accuracy can be much improved by the inclusion of acoustic and mechanical losses, and by allowing for the external acoustic interaction between drive unit and vent. Fig. 6(c) includes the four additional components that are required. From the simplistic viewpoint, R_1 represents box leakage, R_2 the effect of internal air damping material, R_3 the viscous losses in the vent and L_r a first-order correction for the mutual radiation impedance of drive unit and vent. In addition, the value of R_{as} has to be increased to allow for viscous damping caused by the surround of the drive unit.

A useful indicator of realism is the comparison of measured and calculated values for the modulus of electrical input impedance, as shown in Fig. 7; the close match obtained by careful choice of component values can be clearly seen. However, acoustic and mechanical loss or damping cannot in general be represented by a resistance which is constant with frequency. For example, the air leakage from a typical cabinet is likely to be completely negligible, yet R_1 is needed to obtain a reasonable match between calculation and measurement. Values of R_2 or R_3 which varied with frequency in an appropriate manner would probably provide an equally good or better match.

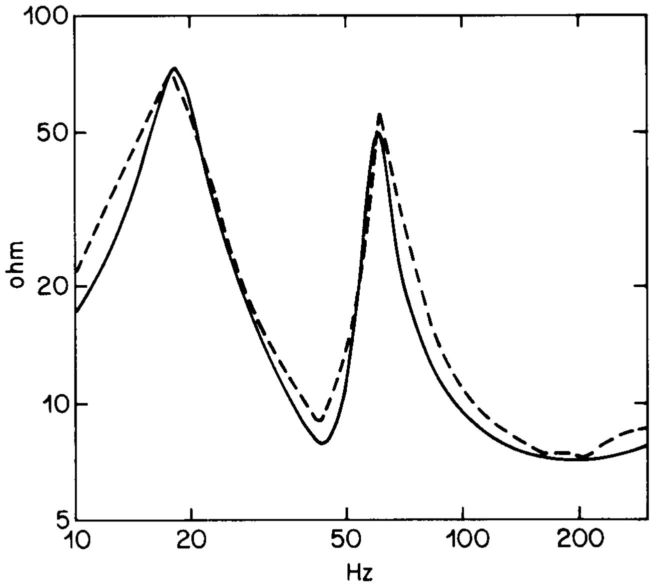

Fig. 7 - Input impedance of LS5/8 bass unit in cabinet.

– – – measured
——— calculated according to model of Fig. 6(c)

Examination of these aspects of the model may well be essential in understanding the detailed behaviour of the system, but for designing a drive unit and box, the simple model is probably most useful, provided that it is regarded as a helpful guide rather than a precision design tool.

5.3 Behaviour at mid and high bass-unit frequencies

The above model suggests that the output of a bass unit in a box should remain flat with increasing frequency until the decreasing wavelength of the radiated sound approaches in dimension the circumference of the diaphragm, i.e. $ka = 1$. Around this frequency, the radiation resistance of the unit ceases to increase as the square of frequency (Eq. 5.2) and asymptotes to the constant value given by[10]

$$R_a = \frac{\rho c}{\pi a^2} \tag{5.5}$$

Comparison with Equation 5.2 shows that this would cause the output to roll off towards higher frequency at -6 dB per octave. In practice, the diaphragm of a typical bass unit ceases to behave as a piston at some frequency below that given by $ka = 1$, so that the effect of Eq. 5.5 is as a rule never observed. Above the frequency of its first mode, the effective mass and radiating area of the diaphragm

appear gradually to decrease in a manner that is certainly beyond simple calculation. The effect is to cause a gentle increase in output with frequency, of the order of 3 dB per octave, until the electrical inductance of the voice coil becomes significant, causing the output to level off. Not far above this, mechanical effects such as modal breakup of the voice-coil former begin to dominate, and the response decreases rapidly.

Fig. 8 shows a response plot for an LS5/9 bass unit, on which the various features just described have been marked.

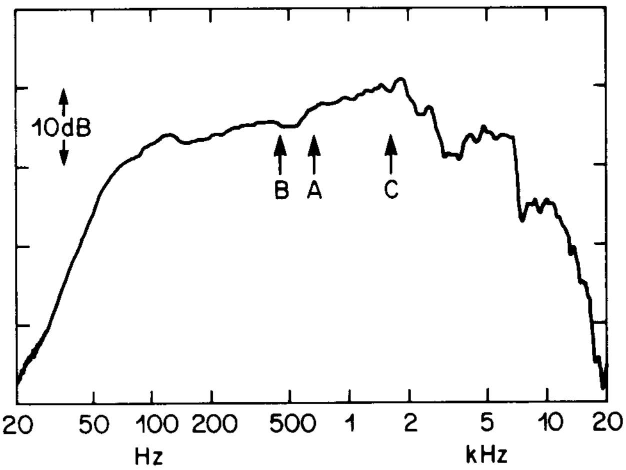

Fig. 8 - Amplitude/frequency response of LS5/9 bass unit in cabinet (free-field measurement on axis), showing effects described in Section 5.3.

A Theoretical upper limit of the model of Fig. 6(c)
B First vibrational mode of cone
C Frequency above which voice-coil inductance dominates

5.4 Structural considerations

The component parts of a typical moving-coil bass driver are shown in Fig. 4(a). Two factors govern their choice: performance and economy. The only two expensive components are the frame or 'basket' and the magnet assembly. Neither of these, at least in the author's experience, has a primary effect on performance provided that the frame is sufficiently rigid to withstand reasonable mechanical shock, and the polepiece dimensions and flux density of the magnet are maintained to acceptable tolerances. For maximum midband efficiency, the flux density should be as high as possible; however, up to about 1.6 T, mild steel polepieces are satisfactory, whereas at higher flux densities special alloys are required, at considerable extra cost. If, as is usual, the voice coil constitutes only between 10% and 20% of the total moving mass, then high motor efficiency may be more economically achieved by increasing the coil diameter than by increasing the flux density, provided that the resultant change in diaphragm shape is acceptable. Both changes require a roughly similar increase in magnetomotive force, and hence in magnet size.

As indicated by Eq. 5.4, the efficiency of a bass unit may be increased by reducing the mass of the moving parts, of which the diaphragm constitutes the major part. The material of which it is made should be light, available in thin sheets, and capable of being moulded into the required shape. No available material is light and stiff enough to form a diaphragm whose lowest vibrational mode is above the unit's usable frequency range; it is therefore desirable that the elastic loss factor be sufficiently high to ensure that modes are well damped. The two more recent BBC monitors, the LS5/8 and LS5/9, have bass-unit diaphragms formed of a propylene-ethylene copolymer of relative density of 0.9, elasticity 1.3×10^9 Pa, and mechanical Q factor around 10. Both objectively and subjectively, it produces better results than any other diaphragm material so far tested here.

The optimum shape for a diaphragm depends on the thickness, density and elasticity of the material, as well as on the properties and dimensions of the surround and voice coil. It must be optimised, in conjunction with these other components, for the best compromise between frequency response, tonal quality, and structural stability. This is a process that so far at least can be carried out only empirically (and very tediously). A good example of the need for compromise was provided by development work on the LS5/9 bass unit. Early prototypes were tonally excellent when first assembled, but the diaphragms gradually deformed at the periphery, where the flare was extremely shallow, with consequent deterioration in tonal and other properties. A change in profile to one with increased peripheral flare completely cured the defect, and the new diaphragms — moulded from the same material — show no perceptible change in any of their properties after about five years. Ironically, the tonal quality has never been quite so good as that achieved initially.

It is traditionally believed that the surround should provide the diaphragm with a good impedance match for transverse waves at its periphery, and be sufficiently lossy that no significant amount of energy is reflected from the frame back to the diaphragm. Laser interferometry shows a mixture of travelling and standing waves at most frequencies, indicating that although some energy is absorbed by the surround, the process is far from perfect. One constructional feature that prevents a smooth impedance transition between diaphragm and surround is the overlap (typically about 5 mm) needed to stick the two materials together; this approximates to a local doubling of sheet density, and thus causes a significant discontinuity in impedance.

Experiments with a variety of surround materials suggest that while the tonal character of a unit is primarily determined by the diaphragm, the use of a particularly unsuitable surround can readily degrade the tonal quality to an extent that renders the unit unusable. The choice of surround material is part of the general empirical optimisation process mentioned earlier. The shape of the surround appears to be much less critical than that of the diaphragm, and usually takes the form of a semicircular 'roll' of sufficient radius to ensure free movement of the diaphragm over the required amplitude of travel. The units mentioned above as having polypropylene diaphragms are fitted with surrounds of plasticised PVC of relative density 1.2, elasticity 1.5×10^8 Pa, and Q of about 2.

Another component which has an appreciable effect on performance, especially at higher frequencies, is the voice-coil former. Usually this is rolled from a thin sheet of high-temperature insulating material such as polyimide to form an almost complete tube; a gap is left to allow for assembly tolerances. In tension or compression along its axis, the coil former is very stiff, although excessive compression will of course lead to buckling. To minimise this risk, and to ensure maximum rigidity of coupling between voice coil and diaphragm, the coil former is usually made as short in the axial direction as production tolerances will permit, thus helping also to to ensure maximum efficiency at the upper extreme of the unit's frequency range. It is possible that the stiffness of the former could be further improved by bridging the gap, where it is not already bridged by the voice coil, with a piece of the same insulating material.

The voice coil must be long enough to accommodate the peak throw of the unit whilst still maintaining a full complement of copper in the magnet gap (unless a short coil is used with a long gap, an arrangement extremely rare in woofer design because it requires an enormous magnet). To avoid problems with the leadout wires, the coil usually consists of an even number of layers of the thickest wire gauge that assembly tolerances will allow. The electrical resistance of the coil would be very coarsely quantised indeed if the only completely free choice were the number of layers (2, 4, 6 etc.), because for constant volume of copper, the resistance is proportional to the fourth power of the number of layers. In practice, the desired resistance is one of the parameters that must be borne in mind when choosing the coil diameter.

Two 'soft' components remain — the dustcap, and the suspension, or 'spider'. Both are usually made of cloth impregnated with phenol-formaldehyde resin, and are available in a variety of forms. The primary purpose of the dustcap is to exclude ferromagnetic particles from the polepiece gap, but it performs an important subsidiary function in acoustically damping the air column contained in the neck of the diaphragm, which otherwise can cause a readily audible coloration.

The fundamental resonance of the unit is determined by the moving mass and the combined stiffness of the surround and spider. The required frequency may usually be obtained by appropriate choice of spider stiffness. Very compliant spiders tend to suffer from long-term creep, so that it is wise to choose the stiffest spider that the required low-frequency performance of the completed loudspeaker assembly will permit.

5.5 Reliability

Loudspeakers may suffer two kinds of damage in use — thermal and mechanical. For broadcast monitoring in particular, it is very important that all possible measures be taken to minimise the risk of failure in service. The most obvious precaution is to ensure that the maximum peak power input is not exceeded; fortunately, in most listening areas, power amplifiers are specified to suit the existing or intended loudspeaker types.

Thermal damage may take two forms — short-term and long-term. The difference can be qualitative as well as quantitative, as can be seen in Fig. 9. This shows an early LS5/8 bass-unit voice coil with a former of unsuitable material. If heated gradually, the former readily withstands the maximum temperature that the coil will ever reach in use. However, the LS5/8 amplifier generates 100 W, and

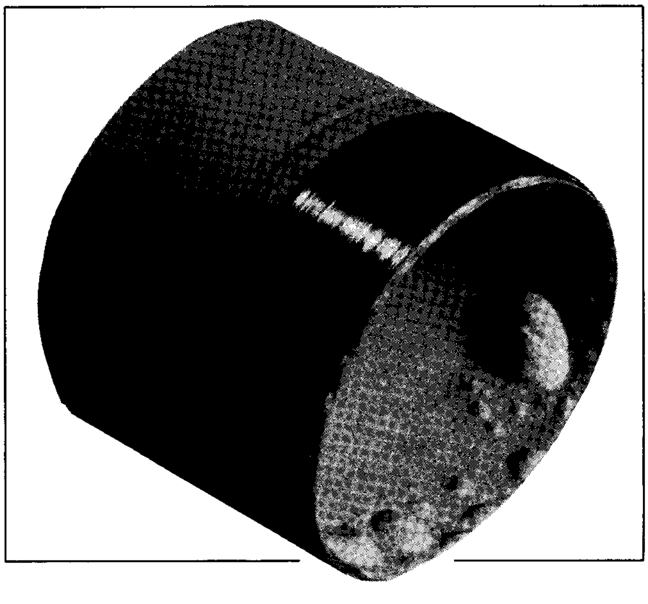

Fig. 9 - Damage to prototype LS5/8 voice coil caused by sudden heating.

the coil contains 6 grammes of copper; the temperature could therefore rise at about 40° per second. Under these conditions, the 'unsuitable' material, which has a layered structure, develops large blisters caused by the sudden formation of vapour between the layers. In fact the insulating material of which the former was made is designed as an insulator for large electrical machines, which, although they may get very hot, do not undergo sudden temperature changes. The use of a homogeneous high-temperature material such as polyimide effects a complete solution to the problem. For this reason the voice coils of the LS5/8 and LS5/9 bass units consist of polyimide-insulated copper wire wound on a polyimide former and held in place by a very high-temperature epoxy resin[4,5].

The voice coil on its former is the component likeliest to suffer thermal damage, whether in the short or long term. As almost all of its heat is transferred to the magnet, a secondary effect of long-term thermal dissipation is an increase in magnet temperature, which in turn heats the spider. This component is not usually designed to withstand prolonged exposure to high temperatures, and if so maltreated, is likely to warp. The peak to RMS ratio of typical audio signals (about 10 dB) is however such that this form of damage happens only if the loudspeaker is subjected to prolonged operation well above clipping level, a rare occurrence even in large broadcasting centres.

Mechanical damage can also be short-term or long-term. If the voice coil is permitted to overshoot its designed range of travel, precautions must be taken to ensure that no damage ensues. The provision, as shown in Fig. 4(a), of a robust 'end-stop' ensures that overload, whether transient or prolonged, does not result in damage to the voice coil by causing it to hit the magnet backplate. One other possible effect of prolonged mechanical overload is fatigue of the leadout wires. Experiments with copper braid show that this has a probable lifetime somewhat shorter than that required for the LS5/8 or LS5/9 bass units. After some search, an alloy braid of copper with 2% beryllium was found, which, as far as can be estimated, is likely to be about 1000 times more resistant to fatigue than copper.

6. CABINET DESIGN

6.1 Introduction

As previously mentioned, the design of a BBC monitor usually begins with box size as the only inflexibly imposed restriction. The relationship between box volume and bass-unit parameters is well described by the lumped-element model, as shown in Section 5; the inertance (acoustic mass) of the air in the vent can be optimised by calculation, followed by some final empirical adjustment in the laboratory.

There are, however, other features of the cabinet that must be taken into account besides its volume and Helmholtz resonance. These and their significance will now be considered in order of increasing audio frequency.

6.2 Vent geometry

A given inertance may be obtained by a tube of almost any cross-sectional area, simply by cutting it to the appropriate length. The maximum required rate of flow in a cabinet vent — which occurs at or near the Helmholtz frequency — must not however lead to excessive air velocity, otherwise turbulence will generate non-linear distortion, in extreme cases causing audible 'puffing' sounds. A useful rule of thumb proposed by Small[21], and with which the author's experience concurs, is that the peak velocity should not exceed about 5% of the speed of sound. Long ago, Thuras[22] et al. provided the basis of more detailed non-linearity calculations. In general, the vent cross-section should be made large rather than small, provided that the pipe length does not exceed about half of the cabinet depth. However, the author has found that vent diameters exceeding about 100 mm can allow noticeable amounts of mid-frequency energy to escape, with consequent curious effects on tonal quality. The extent to which this occurs depends on cabinet size, vent siting, and pipe length.

Usually there are no restrictions on vent shape, and to minimise the effects of turbulence it would appear sensible to make it circular rather than, for example, rectangular. If it is required to handle very high air velocities, then some advantage may be gained by choosing a more 'aerodynamic' nozzle shape than a simple cylindrical pipe.

Because of the low acoustic impedance and small area of the vent, care is needed in the choice of any fabric placed over it if excessive damping is to be avoided. For example, because the cone and surround of an LS5/8 bass unit are of a translucent white colour, it is desirable for cosmetic reasons, and acceptable acoustically, to cover this unit with black voile in addition to the customary woven-nylon grille. Continuation of the voile over the vent, however, was found to have an unacceptable effect on the low-frequency performance of the monitor.

6.3 Cabinet air damping

With increasing frequency, a box ceases to look like a pure 'lumped' compliance and begins to behave more like a small room; the first air mode occurs when the maximum internal dimension equals

one half-wavelength. To avoid tonal colorations and irregularities in frequency response, some form of internal acoustic damping is needed. This is commonly obtained by applying sheets of mineral wool to the internal walls, usually excluding the front baffle; the optimum density of fibre is about 50 kg/m^3. In the LS5/8 box, for example, a standard thickness of 30 mm provides some damping at the first mode (250 Hz) without excessively damping the box at its Helmholtz frequency (45 Hz). The sheets may be wrapped in a layer of polyethylene or similar material to prevent the migration of fibres, provided that it is thin enough to avoid causing excessive reflection at the upper frequency limit of the bass unit, and sufficiently well restrained to avoid audible low-frequency flapping.

Better low-frequency damping could be obtained by suspending the fibre sheet at some distance from the walls, or by filling the whole volume with fibre (presumably excluding some volume around the vent to avoid excessive vent loss). This latter approach would also change the air conditions from approximately adiabatic to approximately isothermal[10], with an effective increase in cabinet volume by a factor of up to 'γ' or 1.42:1. In practice, however, distributed fibre is inconvenient and is rarely used.

6.4 Cabinet structural damping

Most loudspeaker cabinets are still made of plywood or chipboard, with internal wood battens along each edge; they are cheap, strong, and easy to manufacture, provided that complex or curved shapes are not required. Wood, however, is not a mechanically lossy material[23], and can readily exhibit Q factors of 50 to 100; as a result, a typical cabinet has a large number of resonance modes spanning the frequency range of the bass unit that it contains.

The introduction of effective and consistent mechanical damping into any system is rarely achieved without problems; there are several reasons for this. Mechanical loss in solids appears of necessity to be a highly temperature-dependent phenomenon[24], which means in practice that only some specialised solids exhibit high mechanical loss at room temperature; many of these are unstable mixtures which tend to separate into their constituents over long periods of time.

The most effective means of obtaining a stiff heavily-damped sheet is to form a 'sandwich' consisting of a lossy layer between two stiff layers, a technique known as constrained-layer damping. Experiments with cabinets based on this principle have confirmed that the technique could be successful, but have emphasised practical problems. To obtain optimum damping, the shear stiffness of the lossy layer (which is inversely proportional to thickness) must be matched to the longitudinal stiffness of the outer constraining layers (which is proportional to thickness). In practice it is difficult to find lossy materials of sufficiently low shear modulus to be effective with plywood in a constrained-layer system where the sandwiched damping layer should be thin compared to the plywood. This is especially true when it is borne in mind that all three layers must be very well bonded together.

The approach adopted by H.D. Harwood*, and now an established BBC tradition, is to construct cabinets of thin plywood backed by a layer of highly-damped bituminous sheet. This form of construction, like the constrained-layer arrangement, offers very low modal Q factors, but at the expense of requiring a thickness of damping material approximately equal to that of the plywood. However, because of the high density of the 'damping' sheet, the cabinet is stiffness-controlled by the plywood only at low frequencies (typically below 200 Hz); at higher frequencies the mass of the bituminous material is dominant. Construction is simple, and much less critical of inter-layer bonding than constrained damping layers.

Comparison with a variety of commercially obtainable monitoring loudspeakers suggests that the current BBC range is lighter in weight than most proprietary 'equivalents', a very significant consideration in outside-broadcast applications. Extended subjective comparisons show that the audible differences between nominally identical drive units are much greater than those between nominally identical cabinets, suggesting that further economies in cabinet construction may be possible in some cases without impairing loudspeaker consistency, and that these could possibly take the form of a reduction in weight.

6.5 External acoustic effects of the cabinet

Provided that a cabinet is free from high-Q mechanical and acoustic resonances, its size does not critically affect the loudspeaker's tonal quality at mid or high frequencies, at least in the author's experience. However, as mentioned in an earlier section, BBC users are conservative in their choice of box shape, so that only cuboidal boxes have been critically assessed using known drive units.

One property of a loudspeaker assembly that does depend on the size and shape of the box is the directionality at mid and high frequencies. (It is also dependent on the drive units, and of course on the frequency range covered by each unit). Larger boxes, and the larger units that they typically contain, tend to

* Formerly of BBC Research Department.

be appreciably more directional at mid and high frequencies than smaller ones. The general effects of directionality have been discussed in Section 3.

At frequencies above about 2 kHz, acoustic radiation can be markedly affected by details of baffle layout. For example, the LS3/5A monitor has a very small baffle[3], covered by grillecloth at a spacing of about 12 mm. The cloth is supported on a plywood frame, the inside edge of which approaches within 50 mm of the centre of the tweeter dome. To prevent unwanted reflections from the frame edge, which would cause unevenness in the axial frequency response at about 5 kHz, four strips of soft thick felt are mounted on the baffle to form a square surrounding the tweeter.

Another example of the effects of baffle layout is illustrated by the LS5/9. Because this loudspeaker often has to be slung very high due to lack of space in the control area, users had requested that it should have as low a centre of radiation as possible. To achieve this, the vent was not placed below the bass unit in the customary manner; rather, the bass unit was set as low as possible, with the tweeter immediately above it, and the vent in the remaining space to the side of and slightly above the tweeter[4]. It was found, not very surprisingly, that the vent, which has hard reflecting surfaces, could cause slight irregularities in high-frequency response if placed in the baffle at a position very close to the tweeter. The location of the vent was therefore selected with considerable care.

7. THE DESIGN OF HIGH-FREQUENCY DRIVE UNITS

7.1 Introduction

Because the subjective defects that determine the character of a loudspeaker occur chiefly in the frequency range between 300 Hz and 2 kHz, tweeters (unless they are very bad indeed) usually have a less critical effect on perceived tonal quality than bass or mid-frequency units. It may be for this reason that although BBC Research Department has been involved for most of its lifetime with the design of bass-units, its first practical essay into tweeter design took place only about three years ago.

During the past twenty years or so, the general increase in loudspeaker quality has been matched by improvements in commercially available tweeters. Unfortunately, the highest-quality units are not as a rule those most capable of handling the greatly increased power output demanded of modern loudspeakers, and of studio monitors in particular. As a result, the proprietary tweeter selected ten years ago for the LS5/8 monitor was one of very few possible contenders. Although a slightly wider choice exists today, no commercial 'plug-in' replacement appears to be available, and the recent investigation into tweeter design was undertaken to determine whether an acceptable equivalent could be designed, just in case the current unit should ever cease to be available. The detailed results of the work have been described in a separate Report[25], and therefore only the essential conclusions will be summarised here.

7.2 Recent investigations into tweeter design

7.2.1 Choice of unit type

The usual first step in entering any new field of work is to examine the current state of progress. Accordingly, a number of samples of different tweeter types were evaluated, and the results suggested that the moving-coil dome type seemed likely to offer the most practical solution, essentially confirming the conclusions summarised in Section 4.

7.2.2 Considerations of efficiency

Fifteen years ago the spectral energy density of most kinds of BBC programme material was concentrated at low and mid frequencies[26], so that for most monitoring purposes a tweeter was required to generate only a small part (say 10%) of a loudspeaker's total acoustic power output. However, with increasing use of electronic musical instruments in some types of modern and pop music, this fraction has gradually increased, and it is therefore more important than ever that a tweeter should offer high resistance to burnout, which, given a small voice coil and a high power input, is still the chief failure mode of a dynamic tweeter.

Equation 5.4 shows that for maximum efficiency, a dome tweeter should have a large dome, a small moving mass, and a high magnetic flux density. Also, the voice coil wire should have a high ratio of conductivity to density, although the importance of this ratio depends on the mass of conductor as a fraction of the total effective moving mass.

7.2.3 Other design factors

In practice, considerations other than efficiency have to be taken into account. The larger the dome, the more directional is the unit, especially at high frequencies. For directionality, the current 34 mm dome used in the LS5/8 and LS5/9 is probably optimum, but the best proprietary substitute in terms of stiffness (which determines flatness of axial high-frequency response) was found to be a polyester dome

which is 38 mm in diameter, and which, being so large, is excessively directional without treatment.

Whilst the moving mass should ideally be minimised, it was found that the application of a thin coat of plasticised PVC damping compound to the outer 5 mm of the dome and to the suspension roll (formed integrally from the same sheet of material) improved both the subjective quality and the flatness of the measured steady-state amplitude response, as well as slightly reducing the directionality. The tweeter and its component parts are shown in Fig. 4(b).

The magnet of the prototype unit generates a flux density of about 1.5 T, close to the maximum obtainable with mild-steel polepieces.

The conductivity/density ratio of the voice-coil could be doubled by using aluminium instead of copper. By preserving the present ratio of conductor mass to total moving mass (about 70%), the efficiency of the unit could be increased by 3.8 dB, i.e. by 2.4:1. Possible penalties could be an increase in polepiece gap (with a corresponding increase in magnet size if the flux density is to be maintained), and a change in electrical impedance, (which could probably be tolerated by an amplifier that must be powerful enough to drive the bass unit). As the sensitivity and reliability of the prototype tweeters compared well with the units that they were designed to replace, the additional expense and soldering inconvenience of an aluminium coil were not judged to be worthwhile.

It is possible that the power-handling capability of this tweeter could be further increased by the introduction into the magnet gap of a fluid carrying suspended ferromagnetic particles[27], (a 'ferrofluid'). Past work on one-eighth scale model loudspeakers[14] indicates that high shear velocities cause thin fluids to 'spatter' (making an appalling mess) and thick fluids to rupture or cavitate, causing severe non-linear distortion. The model drive unit, however, was required to operate at very high levels over a frequency range from 400 Hz to 10 kHz, whereas a typical tweeter is required to generate levels about 10 dB lower, and only over the range from (say) 2 kHz to 20 kHz. It is possible that by incorporating a suitable ferrofluid into the tweeter design, improvements in power dissipation of up to fivefold could be obtained. It should be pointed out, however, that the maximum temperature of the voice-coil is about 300 °C, whereas that of the fluid is only about 100 °C. The value of the fluid might therefore lie chiefly in reducing the temperature excursions of the voice coil rather than in allowing its power dissipation to be increased; this would reduce the change in sensitivity with programme level and frequency content, an effect known to be perceptible in some high-level monitoring loudspeaker assemblies.

7.3 Summary

Using readily available materials and techniques, it appears possible to design tweeters adequate for use in the current range of BBC studio monitors. The following two points might however be beneficially investigated.

First, the use of slightly more expensive components, such as cobalt-alloy polepieces and aluminium wire, should ensure an increase of about 6 dB in output to set against future requirements. Second, the use of ferrofluid in the magnet gap could reduce thermal changes in unit sensitivity, perhaps rendering the sound balance of multi-unit assemblies less susceptible to changes in drive level.

8. CROSSOVER NETWORKS

8.1 Introduction

The term 'crossover network' is used here to include all of the frequency-splitting and response-shaping filters that a loudspeaker assembly may require. These are usually either small-signal active circuits followed by a separate power amplifier for each drive unit, or else high-level passive circuits which split the output of a single power amplifier into the frequency bands required to drive each unit directly.

Much has been written over many years on the relative merits of active and passive filters, and of the permeable materials (including metal laminations, iron dust, ferrite, and air) which can be used in the inductor cores of passive circuits. More recently, the discussion has extended to include the various types of capacitor (polycarbonate, polypropylene, etc.) usable in both active and passive circuits. The experience of the author may be summarised as follows. Compared to the very audible disparities between nominally identical drive units, the differences between active and passive filters, or between the various types of inductor core, are negligible — *provided that the response/frequency characteristics are the same, and that the amplifiers, as well as the filters and their component inductors, are chosen or designed competently.*

Much has also been written about the importance of achieving phase linearity (or, in more relevant terms, constant group delay) between the input and output of a loudspeaker. The undesirability of severe dispersion (e.g. low frequencies today, high frequencies tomorrow) cannot be doubted, and was indeed well understood in the last century by Heaviside and others. However, the actual perceptibility threshold of differential group delay has never

been investigated in detail, as far as the author is aware, so that no criterion can yet be set. The BBC approach to the matter has always been pragmatic. In comparative tests with variable crossover frequencies, carried out during the development of both the LS5/8 and LS5/9, no coloration coinciding with the crossover frequency was ever reliably detected, provided that care was taken to avoid a peak or dip in the measured steady-state response. In fact, neither the drive units nor the simple second-order filters used in these designs introduce large amounts of group delay. Indeed, in this respect most drive units behave much better than might be expected, approximating closely, at least over their useful frequency range, to minimum-phase devices (a fact first pointed out to the author by L.R. Fincham of KEF Electronics).

8.2 Active versus passive filters

Active filters incur costs in requiring one power amplifier per drive unit rather than one per loudspeaker assembly, whereas passive networks incur costs in requiring specially-wound inductors. Typically, the passive type is slightly cheaper.

The design of active systems is much easier than of passive because each drive unit is buffered by a power amplifier so that its input impedance does not appear among the network design parameters. For the laboratory development of new loudspeaker assemblies, active networks are especially convenient. Also, slight 'dips' to disguise drive-unit colorations, which would increase the number of special inductors in a passive system, may easily and cheaply be incorporated into an active filter.

To make a passive equivalent which is subjectively indistinguishable from its active prototype, the voltage transfer function must be imitated very closely. However, it is rarely possible to say which of two very similar amplitude response curves is actually superior, so that great expenditure of effort in trying to achieve an *exact* equivalent may not be worthwhile.

8.3 Damping factors

There is one other feature of passive crossover networks that may lead to audible differences between apparently similar active and passive filters: namely, the impedance presented to the drive unit, and the consequent degree of electrical damping.

Short-circuiting the voice coil of a drive unit provides a great deal of electromechanical damping, as may readily be observed by tapping the cone of a bass unit in its cabinet, with and without such a short-circuit. A popularly quoted 'damping factor' for power amplifiers is the nominal load resistance (usually 8 Ω) divided by the amplifier's output resistance; this can produce an impressive number often running into thousands. The actual parameter of interest is the degree of damping achieved by the amplifier compared with that obtainable from a short-circuit, the latter being the best that could be done without making special arrangements for conjugate matching of the loudspeaker impedance, which would in reality be complex. Thus defined, the *relative* damping factor for an amplifier of output resistance 0.01 Ω and nominal load resistance 8 Ω would be 8/8.01 = 0.999, more informative than a 'damping factor' of 800.

Similar considerations apply to the interface between a passive filter and a drive unit. Here if we assume that a short-circuit provides a damping factor of unity, then a drive impedance Z_d will provide a relative damping factor Δ given by

$$\Delta = \frac{Z_{ls}}{Z_{ls} + Z_d} \qquad (8.1)$$

The relative damping factor would be zero if the unit were open-circuited, and could approach infinity if perfect conjugate matching were applied to a purely reactive unit. In practice, Δ usually falls between 0.2 and 1.2.

An initial attempt at realising a passive crossover network for the LS5/9 generated a severe coloration at about 800 Hz. After some optimisation to minimise the output impedance of the filter, a considerable improvement in subjective quality level was obtained. The relative damping factors of the bass unit for each filter are plotted in Fig. 10; comparison suggests that the relative damping factor as defined in Equation 8.1 could be an important parameter in passive crossover design, although as yet no criterion can be put forward.

8.4 Frequency shaping or 'equalisation'

The electroacoustic transfer function of a good drive unit is usually reasonably 'smooth', but may exhibit a gradual slope over a wide frequency band. Examples are the bass units of the LS5/8 and LS5/9, whose outputs (in their cabinets) at constant voltage drive increase by 12 dB from 50 Hz to 1 kHz and by 6 dB from 100 Hz to 1 kHz respectively. Frequency shaping of the kind needed to correct this may readily be incorporated into the crossover network, whether active or passive.

Subjectively, a given type of drive unit may exhibit a generic coloration at a reasonably well-defined frequency; attempts at a cure by total redesign may raise more problems than they solve. An example is the LS5/8 bass unit, most specimens of which have a slight but perceptible coloration at about 600 Hz. To

minimise the effect of this coloration, a half-octave wide 'dip' of about 3 dB at 600 Hz has been incorporated into the crossover network. For much the same reason, the LS5/9 circuit includes a similar dip at 1 kHz.

In the past, crossover circuits were often arrived at by trial and error, a process later made easier by measuring instruments with graphical displays or by computer models. (Traditional methods for designing filters with 'flat' passbands can of course be directly applied only where 'built-in' response shaping is not needed). More recently, computer optimisation techniques have been widely applied, including at least one program written for use on a personal computer[28].

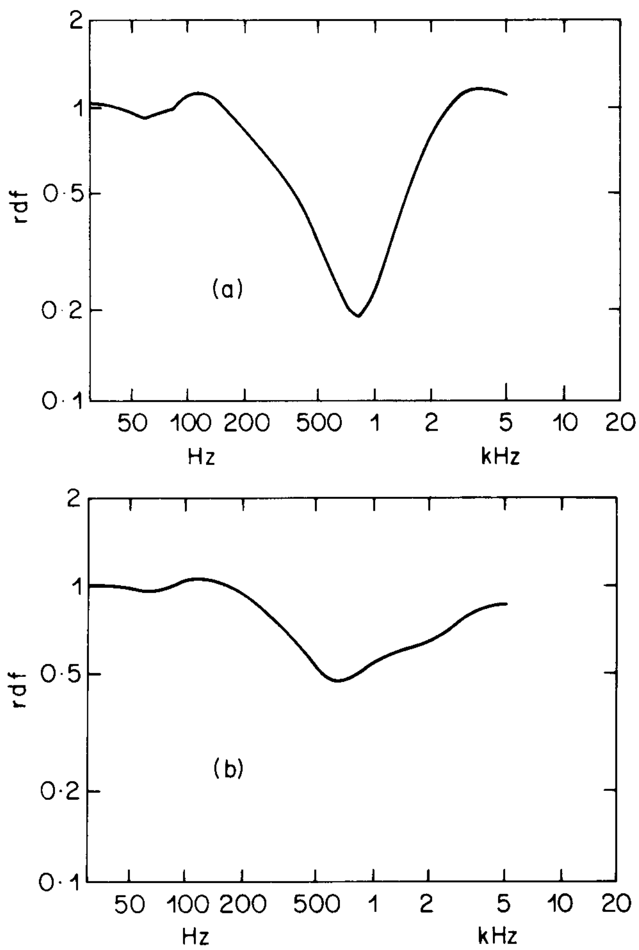

Fig. 10 - LS5/9 passive crossover circuits: low-pass side.

(a) Relative damping factor of first prototype, showing lack of damping at 800 Hz
(b) Relative damping factor of final version.

9. LOUDSPEAKER EVALUATION

9.1 Introduction

The obvious and definitive means of evaluating a loudspeaker is of course by listening to it. An expert listener auditioning known programme material can learn a great deal from a listening test. If all of the sound balancers who use a particular loudspeaker declare it to be excellent, then by definition it *is* excellent. In the author's experience at least, such universal approbation is rare. Although a group of users in an organisation like the BBC usually show remarkable accord in their evaluations, they tend to use adjectives like 'woolly', 'hard', or 'chesty', and nouns like 'honk', 'quack', or 'lisp'. One can often hear what they refer to, but such quirks can rarely be identified by objective measurement, and are very poor guides indeed to any design modifications that might effect significant tonal improvements. (Very rarely, complimentary expressions like 'clean' or 'uncoloured' are applied; perhaps one reason for the rarity of these is that a perfect loudspeaker should presumably have no perceptible characteristics of its own.)

What is required, of course, is a well-defined relationship between subjective peculiarities, measurable deviations from 'ideal' acoustic output, and oddities in physical behaviour. A 'dreadful quack at 800 Hz' should be confirmed by a disturbance in the otherwise serene acoustic time-frequency-acceptability plot, and by an agonised writhing at 800 Hz to disturb the otherwise exemplary piston-like movement of the diaphragm.

Reality is otherwise. 'Good' loudspeaker drive units appear to exhibit just as complex mechanical and acoustic behaviour as 'bad' ones. The author is currently engaged in a project to try to find *some* relationship between the subjective, acoustic, and mechanical facets of loudspeaker behaviour. This has been undertaken in the knowledge that previous attempts during four decades have not yielded a final solution. Results (positive or negative) will be published in due course. Two reference works only are listed relating to this subject[29,30]; each includes an extensive bibliography.

9.2 Subjective evaluation

Experience shows that comparative judgements of loudspeaker quality can be made more consistently than absolute ones. An absolute assessment of a new design is something which emerges gradually out of weeks or months of use in control rooms. Often, a pair of new loudspeakers sent out for 'field trial' will be received with cautious approval, yet returned after a month or two with a list of criticisms detailing points that have emerged only gradually from continuous use.

For comparative tests, a reference loudspeaker is of course needed. This is provisionally selected during the early stages of commercial production as

being a typical unit of acceptable quality; once production is well established, a new reference may be adopted as a clearer picture emerges of what is 'typical'. In fact, at least three such units are selected in normal BBC practice, to provide a working standard for acceptance testing: a spare (which is carefully stored): and a standard by which the manufacturers can assess the consistency of their output, whether by listening or by measurement. An established standard is also of course the only reasonable reference available in appraising a new design.

In listening tests, it is important that the listener should begin with as few preconceived ideas as possible. For example, a look at a response plot may cause him, consciously or otherwise, to listen for some expected peculiarities. Normally, an A/B switch is provided, and the loudspeaker to be used as reference is indicated. The loudspeakers are placed behind an acoustically transparent but optically opaque curtain, especially if any aspect of the units under test might be visually identifiable. To help eliminate room effects, the test may be repeated with the loudspeaker positions interchanged. If several units are to be tested, it is useful to include one twice — anonymously — to test the listener's consistency. (Experienced listeners expect this.)

Finally, it is essential that the listener delivers his judgement before any additional information is given to him; not (one would trust) that he might 'cheat', but rather that he might re-interpret what he thought he had heard in the light of further knowledge. Subsequent discussion may well prove valuable, but must be *subsequent.*

Formal tests involving a number of listeners may need further care, particularly if, as is likely, they permit less in the way of personal communication between subjects and test organiser. Past experience suggests that a particular hazard is the use of descriptive terms whose meaning seems obvious to everyone, but which can actually mean different things to different people.

9.3 Measurements of acoustic output

It has long been recognised that reasonable flatness of steady-state amplitude response, measured on axis in free field, is a necessary but not sufficient requirement for a high-quality loudspeaker. A three-dimensional isometric time-frequency-amplitude plot, or 'spectrogram' was first suggested in 1946[31] as a means of characterising a loudspeaker, but remained of mainly academic interest until about 1973 when digital technology and the fast Fourier transform made its use a practical proposition[32]. Recent years have seen the computation of the Wigner distribution[33,34], a function of time and frequency whose magnitude approximately corresponds to energy. Some investigations of the Wigner distribution have also been carried out here at BBC Research Department. Like the spectrogram, it may be represented isometrically, or in plan view as either a contour plot or a greyscale density map. In the author's experience to date, neither the spectrogram nor the Wigner distribution has been found easy to relate to subjective findings.

Less esoteric parameters are directionality and amplitude non-linearity. Directionality, as pointed out in Section 3, influences the relationship between a loudspeaker and its environment, but has no consistent effect on perceived tonality. Although directionality is far from being unimportant, it probably contributes very little to the 'mystery' of why loudspeakers sound as they do. Nor is non-linearity a primary factor; if it were, then tonal quality would be highly dependent on drive level, which, within a very wide range, it manifestly is not.

9.4 Measurements of diaphragm movement

The use of laser interferometry to examine loudspeaker diaphragm behaviour is by now well established[35], and has been adopted by many loudspeaker researchers, including the BBC[36]. The technique has made available for the first time a detailed picture of the movement at all points on the diaphragm at any frequency; herein lies one of the minor problems in its use. Two dimensions are needed to represent position on the diaphragm, and the measured velocity (or displacement) requires a third. A complete representation of diaphragm movement would also include frequency as a fourth dimension. In practice, modal frequencies are readily identified by performing frequency sweeps at two or three fixed points on the diaphragm; perhaps not surprisingly, a large number of such frequencies can usually be found.

One practical disadvantage of the technique is that to obtain high reflectivity, the surface of the diaphragm must be coated with magnesium oxide by holding it over burning magnesium. With care, heat damage can be avoided, but the coating obtained can significantly increase the mass of the diaphragm (e.g. by up to 10% for a typical tweeter).

In recent BBC-designed bass units, interferometry has proved very valuable in showing up a variety of potential problems such as low-frequency diaphragm modes (much damped by the choice of suitable surround material and thickness) and dustcap 'rocking' modes (eliminated by a change of dustcap type and greater care in its symmetrical placement).

9.5 Relating objective measurement to subjective judgement

The previous three sub-sections may have conveyed some indications of why it is not easy to relate subjective and objective assessments of loudspeakers. On the objective side, powerful and accurate techniques are available which are capable of yielding an enormous amount of data. On the subjective side, judgements are usually comparative rather than absolute; they are inevitably imprecise in nature, difficult to make, and even more difficult to describe in terms that mean the same thing to all listeners.

In fact it is in the subjective field that progress is most needed, to provide data which are statistically verifiable and at least partially quantitative in nature. As mentioned above, a project is currently under way which it is hoped may constitute a step in this direction.

10. CONCLUSION

Loudspeaker design emerged perceptibly as a branch of acoustics at some time around the beginning of this century, partly as an alliance between art and science, and partly as a conflict. Vestiges of the conflict remain to this day. Gradually, however, the proportion of science in the mixture has increased, as a result of the contributions, all over the world, of manufacturers, academics, and broadcasters, as well as those who are simply enthusiasts. The references cited in the next section include (at least in the author's view) a representative selection of these contributions. The primary purpose of this Report has not, however, been to review the literature, but rather to consider the subject from the viewpoint of an organisation which has maintained a 'presence' in the field for more than forty years. Descriptions of current BBC loudspeaker designs have been published in separate Reports[3,4,5]; these are referred to but are not quoted in detail in this present one.

Acoustic theory is very important in trying to understand loudspeaker behaviour. Nevertheless, the BBC's approach has always been pragmatic, and where theory and practice disagree, then the theory has always been assumed to be wrong. Broadcasters who design their own loudspeakers are fortunate in having a large number of skilled users who are not party to the design process, and whose opinions are therefore not prejudiced by beliefs about how a particular loudspeaker should behave. These opinions, however, although easy to canvass, are very difficult to quantify, and efforts to relate them more closely to objective measurements still continue.

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An excellent general work has recently appeared, entitled 'Loudspeakers', edited by J. Borwick and published by Butterworth & Co., 1988.